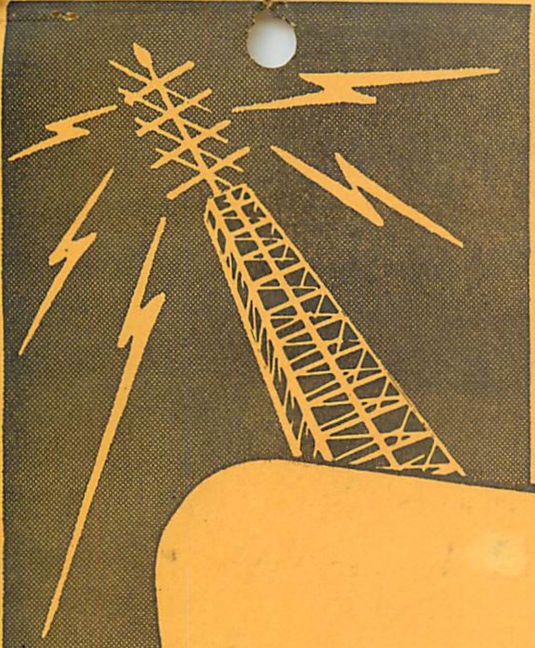
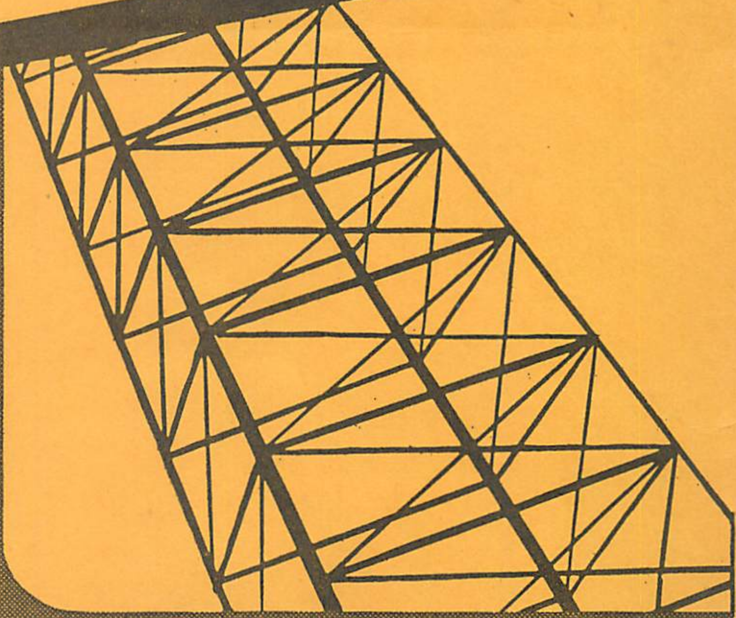


LESSON FTR - 19

SOUND SYSTEMS



DE FOREST'S TRAINING, INC.

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LESSON FTR-19

SOUND SYSTEMS

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Learning without thought is labor lost. Thought without learning is intellectual death.

-- Confucius, 550 B.C.

PUBLIC ADDRESS SYSTEMS

The earliest models of microphones and headphones were designed for the original telephone systems which had a comparatively short operating range over wired circuits. Many devices were developed to extend this range and shortly after the invention of the triode tube, the telephone companies built the first "electronic repeaters" which now are known generally as audio amplifiers.

With the advent of Broadcast Radio, telephone types of microphones were used for transmission and headphones for reception to make up what can be thought of as a Wireless telephone system. However, the requirements of radio differ quite widely from those of a telephone. At the transmitter, the microphone must be more sensitive to pick up sounds over a larger area and must respond to a much wider band of frequencies than required for speech only. At the receiver, the signals must be reproduced at a volume level high enough to be heard easily and distinctly without holding a headphone to the ear.

With the rapid growth of radio, the original telephone units were modified to meet the new requirements and naturally, new and improved types were developed. Then, it became evident that if the audio amplifier of a radio receiver could raise the level of the weak signals of the detector output it could do the same for the output of a microphone connected directly to it. Thus, by the use of a microphone, audio amplifier and one or more speakers, the voice of a person can be reinforced and increased to almost any desired volume level. Used mainly at large gatherings, arrangements of this type are known usually as Public Address Systems.

In general, a public address system includes one or more microphones and other pickup devices, one or more audio amplifiers and one or more speakers each capable of handling considerable power. There are many specific applications for equipment of this type which today will be found in practically all auditoriums, churches, clubs, ballrooms, baseball parks and other similar places which accommodate audiences large or small.

SELECTION OF EQUIPMENT

In the selection of equipment for any sound system, the size of the installation, the use to which it is to be put, the quality desired, and the average noise level at the location must all be considered. The type of system used for a large outdoor audience at a race track or baseball park, where there is a large amount of audience noise, is quite different from that used for providing background dinner music in a restaurant, or for sound reinforcement in a concert hall.

These factors determine the power requirements, fidelity, amplifier gain, controls, input sources, loudspeakers, and transmission lines. For outdoor gatherings such as fairs and picnics, where the size of the group may range anywhere from about 100 to 25,000 people, and where the noise level is high, a portable system having sufficient power for adequate coverage and intelligibility is needed.

For an open air concert, where the audience may number from 500 to 5000 people, the noise level is not high, and a medium power system capable of high quality reproduction is needed. For a restaurant, where 50 to 300 people may be seated, the noise level is medium and a fairly low power system having a mellow quality is desirable. For a dance hall the system should have a good bass response.

The desired characteristics of a system are obtained by the proper selection of parts which can be divided into the three main divisions of (1) Input devices, such as microphones, phono pickups, and radio tuners, (2) The amplifier and (3) The number and types of speakers. Amplifiers, loudspeakers, microphones and phonograph pickups have been described in previous assignments. However, in respect to their selection for use in p-a systems, the following review of their characteristics will be of benefit.

MICROPHONES

Although the output of the carbon microphones is relatively high, its response at the higher audio frequencies is low, therefore it is not generally employed for the pickup of music or other programs which require high fidelity reproduction. However, it does find application where speech frequencies only are involved, such as in police, amateur, and aircraft radio telephone systems. Other disadvantages of the carbon microphone are its high noise level due to the great number of contacts presented by the large number of carbon granules, a tendency of the granules to pack together, and the effect of vibration, positioning, handling, etc. on its sensitivity. Its advantages are its comparative ruggedness and low cost.

The condenser microphone has a better high frequency response characteristic with a much lower noise level than the carbon microphone, and is not affected appreciably by temperature, humidity, or ordinary handling. However, it is affected by cavity, diaphragm, and air-pocket resonances and its sensitivity is much lower than that of the carbon type. Because of this low sensitivity, a preamplifier is usually mounted in the same case or housing along with the microphone itself.

From the viewpoint of the sound engineer, this arrangement constitutes a disadvantage because preamplifier noises and failures are often encountered while the microphone is in use. It then becomes necessary to disconnect the microphone from its preamplifier and connect it to a spare while a program is in progress. These interruptions are highly undesirable and the condenser microphone is seldom encountered today except in special applications such as in laboratory test equipment.

The dynamic microphone is popular for p-a systems and for remote pickups as well as studio work in radio because of its adaptability. Due to its low impedance, it may be used with shielded cable over considerable distances from its preamplifier and it has the advantage of mechanical ruggedness with a higher sensitivity than the condenser type. Most dynamic microphones are capable of operating with uniform frequency response up to about seven or eight thousand cps and their noise output is comparatively low.

A uniform frequency response up to about 15,000 cps can be obtained by use of a crystal microphone, which has low noise output, and is relatively inexpensive. The advantage of its high impedance is that the microphone output can be fed directly into the grid of an amplifier tube without the necessity of an impedance matching transformer. However, the sensitivity of the crystal type is comparatively low and therefore it should not be located at too great a distance from its preamplifier. The crystal microphone is non-directional, that is, it picks up sound equally well from all directions and the desirability of this characteristic should be considered in the selection of a microphone for a particular application.

Next to the crystal in its quality of frequency response characteristic is the ribbon, or velocity, microphone. Its sensitivity is low, but its impedance is low also and therefore it may be connected to its preamplifier by means of a comparatively long shielded low impedance transmission line. That is, the preamplifier need not be close to the ribbon microphone as with the condenser and crystal types. However, it may be easily overloaded and sudden sounds such as gun shots, explosions, etc., may blow the ribbon entirely out of the air gap. For this reason, a different type, such as a dynamic, is usually used for sound effects in radio studio work.

The ribbon microphone is bidirectional in that it is most responsive to sounds originating at its front and rear, but least responsive to sounds originating at its sides. This characteristic is of considerable value in the solution of some of the difficulties usually encountered in reverberant locations, because it reduces the effect of undesired sound

reflections. Also, it increases the possibilities of obtaining better balance, clarity, naturalness and selectivity. When used for public address purposes, the directional characteristic of the ribbon microphone permits the reduction of feedback effects between the microphone and loudspeaker.

MICROPHONE PLACEMENT

Possible ribbon microphone placements illustrating the use of its directional chracteristics are given in the diagrams of Figure 1. In any particular direction the sensitivity is directly proportional to the length of a line drawn from the microphone to the point where it intersects one of the circles.

Figure 1A shows the general arrangement for a soloist and piano. The actual distance must be determined by the strength of his or her voice, but should not be less than 2 feet. As shown by the dotted-line piano, interesting effects may be obtained by changing the angle of the piano with respect to the microphone, thus changing the ratio of reverberation to direct pickup.

For stage plays, movements of the actors seeking advantageous positions may be avoided by utilizing its directional characteristics as shown in the arrangement of Figure 1B. If the microphone is used by a speaker seated at a table, it should be placed so that it picks up sound directly from the speaker rather than that reflected from the surface of the table.

A studio arrangement of a ribbon microphone, for soloist and orchestra is given in Figure 1C. Because artists and announcers cannot work closer than two (and preferably three) feet from the microphone, difficulty is sometimes experienced in obtaining the proper balance between the artist or announcer and the orchestra. This difficulty may be overcome by using two microphones, one for the orchestra and one for the artist or announcer. When this is done, the artist's microphone should be oriented so that its dead zone is toward the orchestra.

In public address work, usually the microphone can be placed within three or four feet of the speaker, but to prevent acoustic feedback it should be arranged so that its direction of minimum pickup is toward the loudspeaker system. In certain cases it may be necessary to have a microphone installed at each side of the stage to give the speaker the required freedom of movement.

By unusual microphone construction, the bidirectional ribbon characteristics of Figure 2A can be combined with the nondirectional dynamic microphone characteristics of Figure 2B to provide the cardioid shape of Figure 2C. For this reason the microphone is called unidirectional, and is useful where it is necessary to discriminate against sounds coming from the rear which would be picked up by the bidirectional type.

PHONO PICKUPS

In a complete p-a system, it is desirable and often necessary to make provisions for the reproduction of phonograph records either to provide musical programs or supplement the voice and music of live performers. For this service, it is common practice to install a phonograph turntable and pickup the output of which is similar in frequency but of higher amplitude than that of most microphones.

As previously explained, many types of phono pickups have been developed, but where cost is an important factor the best performance will be obtained from the crystal-type like those generally found in the better home type phonograph equipment designed for 10 inch or 12 inch records. High fidelity magnetic and dynamic pickups are more expensive, but have a wider frequency range with lower distortion and are used mainly with equipment designed for the professional type of 16 inch transcriptions.

In many cases it is desirable that arrangements be made so that the p-a system can reproduce radio programs as they are broadcast. For this purpose there are special types of radio receivers, called radio tuners, which include all the high frequency tuned circuits and terminate at the detector. As far as the p-a amplifier is concerned, this is merely another type of input and as the amplitude of its output compares quite closely to that of a phono pickup, usually it is connected to the phono input circuits of the amplifier. Some commercial p-a amplifiers provide a connection specifically designated as the radio input.

Thus, a complete p-a amplifier may make provision for three types of input signals, (1) Microphone, (2) Phono and (3) Radio. These are classed according to the source of the signals because the frequencies of all of them are in the audio band. The amplifiers of the larger sound systems may include input circuits for one, two or more of each type of input with the necessary controls for each.

P-A AMPLIFIERS

The explanations of the earlier lessons described the various separate circuits of the main types of audio amplifiers and controls but, for a brief review, we show them all combined in Figure 3 and will trace the paths of the signals from the input circuits to the output transformer.

Starting with input circuit of MIC. #2, the output of the microphone develops a voltage across the grid load R2 and thus applies the signal to the control grid of V2. The bias voltage

for this grid is obtained from the drop across R3 which carries the cathode currents of V1 and V2. The signal voltage on the control grid of V2 causes variations of plate current which, in turn, vary the voltage across the load resistance R10. These voltage changes are carried over to the left hand grid of V3 through coupling condenser C3 and potentiometer R12 which acts as the grid resistor and also as a volume control for this channel.

The plate voltage for V2 is obtained from a tap on the voltage divider made up of resistances R11, R5 and R4. The screen-grid voltage is also obtained from this divider, but is tapped off between R5 and R4, and thus the screen operates at a lower potential than the plate. The bias voltage for the left hand grid of V3 is obtained from the voltage drop across R13.

With the signal applied to the left hand grid of V3, there will be a variation of current in the plate load resistance R15, and the voltage drop across it is carried over to the control grid of V4 through the coupling condenser and grid load R18. This coupling condenser is made up of C8 and C9 so that the total reactance can be controlled to a great extent by varying the position of the contact of R17 which acts as a tone control. The screen grid and plate of the pentode V4 are connected together so that the tube operates as a triode.

The plate circuit of tube V4 is coupled to the grid circuits of the output tubes, V5 and V6, by means of transformer T1. Notice here, that instead of the usual center tap for push pull operation, the secondary consists of two separate windings to permit inverse feedback action by means of the voltage drops across resistors R22 and R23. The plate circuits of the output tubes are coupled to the speakers by output transformer T2 which has a center tapped primary and two secondaries, each of which has several taps to provide various output impedances for matching different combinations of speakers.

When the output voltage of a microphone is applied to the MIC. #1 input channel, it will appear across R1 and be applied to the grid of tube V1. Like V2, this tube obtains plate and screen grid voltages from a divider arrangement consisting of R11, R7 and R6 and its grid bias from the common cathode resistor R3, with its bypass condenser C. Variations of voltage on the control grid cause changes of current in the load resistor R8.

The resulting variations in voltage drop are applied to the right hand grid of V3 through the coupling condenser C1, and the tapped potentiometer R9. This, of course, assumes that the movable arm of R9 is in its present position or somewhere

between grounded center tap and the condenser C1 end. By changing the position of the arm of potentiometer R9, between the center tap and the coupling condenser C1 end, the amount of signal voltage from this channel can be controlled. Because the plates of V3 are connected directly to each other a signal on the right hand grid of V3, is carried on through the succeeding stages the same as explained for channel #2.

If the output of a phonograph pickup is applied across the phono input connections, there will be a corresponding voltage drop between the upper end of R9 and the grounded center tap. Therefore, to apply this signal to the right hand grid of V3, it will be necessary to set the movable contact of R9 somewhere between its upper end and the center tap. The phono pickup feeds directly into the second stage of V3 because its output voltage is considerably higher than that of a low-level microphone, and therefore requires less amplification.

With the movable arm of R9 in the position shown, the input signal to the right grid of V3 will be from the MIC. #1 channel, and when the arm is moved into the upper position, the input signal will be from the phono input. Thus, R9 not only controls the volume of these two channels, but it acts also as a fader so that the signal from either channel can be applied to the right-hand grid of V3. In changing from one channel to the other there is no sudden cut-off, but when moving the arm from one end to the other, after the volume of one input signal is reduced to zero, the volume of the other starts to increase. It is because of this gradual change from one signal source to another, that a control like R9 is commonly referred to as a Fader.

We will assume that a pickup is connected across the phono input and the fader is in a position to apply the signal to the right-hand grid of V3. Also, we will assume that a microphone is connected to the MIC. #2 channel, so that its output will be applied to the left grid of V3 through V2 and the coupling network.

Under these conditions, there will be two separate and distinct signal sources applied to the respective grids of V3. However, because the plates of V3 are tied together, there can be but one value of signal current which will have a waveform dependent on the values of the two signals. In other words, the signals will mix, and because this has been accomplished electronically, a circuit of this kind is known as an Electronic Mixer. Notice that both the phono and MIC. #1 channels can be mixed with MIC. #2 channel.

Thus control R9 serves as a fader for the phono and MIC. #1 channels as well as a volume control for each while control

R12 serves only as a volume control for the MIC. #2 channel. Operated together, these two controls serve as a mixer for signals in MIC. #2 channel and those in either the phono or MIC. #1 channels.

Possibly you are wondering about the advantage of such an arrangement, and a good example is a piano accompaniment for a vocalist. The usual procedure is to place one microphone close to the piano and the other in a position so that the vocalist can be more or less at a distance from the accompanying music as indicated in Figure 1A. By properly manipulating the individual channel volume controls, the most desired sound levels of each can be obtained. Without a mixer, it would be necessary to use only one microphone; and it would be impossible to regulate the two signals separately in order to obtain the desired relative levels.

MIXER CIRCUITS

In addition to the mixer circuits shown by the schematic diagram of Figure 3, another simple arrangement employing two potentiometers is that shown by Figure 4A. Here, when the sliding contact of potentiometer R1 is moved to the lower, or ground end, there will be no signal input from source A to the amplifier tube V1 and the input from source B is regulated with potentiometer R2. When the slider on R2 is moved to its lower end, there will be no input from source B, and the input from source A may be adjusted with R1. If inputs from both A and B are desired, their signal levels may be adjusted with their respective controls R1 and R2. However, this arrangement has a disadvantage of hum pickup because both sides of channel B are above ground. Also its stray capacitances to ground tend to cause attenuation of the high frequencies of channel A.

A more satisfactory mixer arrangement, similar to that of Figure 3 and shown in Figure 4B, is to feed a single input to each grid of two tubes having a common plate load. This method results in isolating the input circuits sufficiently so that the setting of one volume control has no effect on the other. Here, the signal from source A, appearing across R1, is applied to the control grid of tube V1, causing variations in its plate current. Also, the signal from source B, appearing across R2, is applied to the control grid of tube V2 resulting in similar variations in its plate current. Since the plate currents of both tubes pass through the common load resistor R5, the voltage variations across this resistor will correspond to the signals from both sources and will be coupled through condenser C3 to the following tube of an audio amplifier. The linearity of this mixer circuit can be improved considerably but with reduced gain by the omission of the cathode bypass condenser C1.

In order that similar settings of the controls will produce similar output voltages, in any mixing system it is desirable that the input voltages to all channels be as nearly equal as possible. Therefore, when the output of a low-level microphone is to be mixed with that from a phono pickup, usually it is necessary to incorporate one stage to raise the microphone output level to that of the phono pickup.

One such arrangement is employed in the input circuits of the preamplifier of Figure 3, where tubes V1 and V2 raise the signal levels of MIC. #1 and MIC. #2, respectively, and the outputs of both of these tubes are mixed in the V3 stage with that of the phono pickup.

A somewhat larger mixing system which provides adequate control of two microphones and two pickups is shown in Figure 4C. The input signals from the phono pickups are controlled with potentiometers R3 and R4, respectively, which are connected in parallel with resistors R6 and R7 connected in series with the grid at V4 to prevent the potentiometers shorting out one another.

The microphone outputs of Figure 4C are impressed separately on the grids of a double triode tube V2, thus providing complete isolation. Switch S1 provides a signal circuit from either microphone amplifier into tube V3 of the mixing stage. The input of V3 is controlled by means of the potentiometer R10 and isolation is provided by feeding the phono signals into tube V4. Finally, all the signals are combined across the common plate load resistor R13, and coupled to the following amplifier stage through condenser C5. The remaining components in this partial circuit are convention and thus their functions will not be repeated here.

VOLUME CONTROLS

So far we have given you a general explanation of the simple types of volume controls used in a-f systems. However, there are many different types of controls and circuits and, because of their importance, we are going to describe a few of them at this time.

Starting with Figure 5A, we show the circuits of a magnetic type phonograph pickup P connected to the input of a triode amplifier tube V. The pickup generates a voltage and here, the entire resistance R is connected to serve as a load. With a complete circuit through P and R, the generated voltage will cause a current in R which, in turn, will cause a voltage drop.

The current will vary with the generated voltage and the voltage drop across R will vary with the current in it. In this way, the voltage generated in the pickup coil will appear across across the resistance. By making resistance R of the potentiometer type and connecting the grid return to the ground end and the control grid of V to the movable contact, a simple volume control is provided.

According to Ohm's Law, voltage drop is equal to the current times the resistance and moving the arm of the potentiometer varies the amount of resistance in the grid circuit of the tube. Thus, by moving the arm, it is possible to vary the grid voltage from 0 to the full drop across the entire resistance. Plate current through resistance R1 provides the necessary grid bias voltage for normal operation of the tube.

For Figure 5B, we again have the units of Figure 5A, but the connections are changed so that the pickup connects from one end of the potentiometer to the moving arm, while the entire resistance is across the grid circuit at all times. As the arm is moved, the value of resistance across the pickup is varied, thus changing both the current and voltage drop.

The grid circuit consists of the entire resistance R, the grid cathode resistance and bias resistor R1. As the grid circuit carries no current under usual operating conditions, there will be no voltage drop across that part of the potentiometer resistance which does not carry the pickup current. However, plate current in R1 causes the voltage drop for the grid bias.

For the circuit of Figure 5A, the entire resistance of the potentiometer is connected as the load on the pickup, while for the circuit of Figure 5B, the entire resistance of the potentiometer is always in the grid circuit but, as the movable contact is adjusted for different volume levels, the load on the pickup will be changed.

This variation of either input or output impedance, especially at low volume settings, may cause changes in the frequency response of the stage and therefore, a control of this kind would not be suitable in circuits which require a constant impedance. However, you will find this type of control commonly used both in radio receivers and p-a amplifiers because the variations of frequency response, caused by changes of volume setting, are not sufficient to be objectionable except where high fidelity reproduction is desired.

L TYPE VOLUME CONTROL

Many of the troubles caused by changes of load in grid circuits are overcome by the L type slide wire volume control of

Figure 5C that includes two separate resistances, R1 and R2, with a common slider which makes contact with both. Conventionally, plate current in R3 provides the grid bias for the tube.

Checking over this circuit, you will notice R1 is permanently connected across the pickup P the same as the potentiometer of Figure 5A. Instead of connecting directly to the grid of the tube V, the slide also operates on the other resistance R2 which is in series with the grid. As the volume is reduced by moving contact A toward the lower end of R1, contact B moves toward the lower end of R2. Thus, as that part of R1 in the grid circuit is reduced, this loss of resistance is compensated by adding more of R2.

When properly designed, a volume control of this type will maintain a constant impedance across the pickup and also a constant impedance across the grid, or input circuit, of the tube. It has been used with very good results both as input and volume control for individual speakers and headphones.

T TYPE VOLUME CONTROL

An improved type of constant input and output impedance volume control is the T arrangement of Figure 5D which has three separate resistances, R1, R2 and R3, each with a sliding contact represented by the double pointed arrow "B". All three sliding contacts are connected electrically and move together.

With slider B moved to the top of Figure 5D, resistance R2 is shunted across both the pickup and transformer primary while R1 and R3 are not active in the circuit. For best results, the value of R2, the impedance of the pickup P1, and the primary of the transformer should be equal, and as we will explain later, resistances R1 and R3 should have a value equal to that of R2.

Starting with slider B at the top, there is the maximum coupling between the pickup and transformer primary because the maximum pickup voltage is across them both. As the bar is moved down, a part of resistance R1 is added in series with the pickup, and part of resistance R3 is placed in series with the transformer primary. As these resistances are equal and the sliding contacts are fastened together, an equal amount of resistance has been inserted into each circuit. At the same time, an equal amount of resistance R2 has been cut out of the circuit, therefore, while the value of resistance across the pickup or transformer primary has not been changed, the coupling between them has.

For example, suppose the pickup and transformer primary each have an impedance of 500 ohms. As we explained before, R1, R2 and R3 will also have a value of 500 ohms each. With the contact arm set exactly in the center, as shown in Figure 5D, across the pickup there will be 250 ohms of R1 plus 250 ohms of R2, for a total of 500 ohms. Looking at the output side of the circuit, there will be 250 ohms of R3 and 250 ohms of R2 for a total of 500 ohms across the transformer primary. With the contact arm four fifths of the way down, there will be 400 ohms of R1 and 100 ohms of R2 for a total of 500 ohms across the transformer primary.

Suppose now the pickup produces 5 volts. With the bar all the way up, all this voltage will appear across R2 and also the transformer primary. With the bar half way down, part of this 5 volts will be lost by the drop across that part of R1 which is in series with the pickup. The remaining voltage across R2 will not all be applied across the transformer primary because there will be a drop across that part of R3 in series with the transformer. Thus, without changing the value of resistance across either the input or output circuits, adjustment of bar B controls the output. It is because the impedances are matched at all times that the T type of volume control does not produce any noticeable attenuation of any frequency at any setting.

TONE CONTROL CIRCUITS

The action of series and parallel circuits containing inductance, capacitance and resistance are used in many ways for controlling or improving the frequency response of audio amplifiers. For example, a low pass filter will reduce the higher frequencies and thus emphasize the lows while a high pass filter will reduce the lows and emphasize the highs. By making either or both of these adjustable, we have a Tone Control.

Figure 6A shows the circuit of one of the simplest and perhaps most common type of tone control. The complete circuit comprises a resistance coupled audio amplifier stage in which the resistor R serves as the grid load, R1 as the bias resistor and R3 as the plate load. Without considering C and R2, we will assume that the frequency response of this stage is perfectly flat. That is, with constant input voltage both high and low frequency voltages appear across the plate load R3 with equal amplitude.

Now, assume that C and R2 are connected in the circuit with the movable contact at the upper end of R2. In effect, this removes the resistance and places the condenser C between the plate of tube V and ground. As the reactance of a condenser

varies inversely as the frequency, it will provide a comparatively low reactance path for the high frequencies thereby reducing the voltage drop across the plate load resistor R3. Under these conditions, the frequency response of the stage is no longer flat and the higher frequencies will be attenuated by an amount depending upon the value of C.

As it is impractical to construct a variable condenser of the required capacitance, by changing the value of resistance in series with C, the total impedance of the circuit is varied and thus the high frequency attenuation can be controlled. The series tone control arrangement of Figure 6A is sometimes employed in the control grid circuit, with the condenser C and resistance R2 connected in series from the grid to ground. The action however, is exactly the same as explained above.

The circuit of Figure 6A will attenuate only the high frequencies, and to attenuate either the highs or lows, the circuit of Figure 6B is sometimes employed. Notice that the amplifier stage is exactly the same as that in Figure 6A while the tone control made up of the condensers C, C1 and C2 together with the variable resistors R2 and R4 is like that shown in Figure 3. The variable resistances are "ganged", or controlled by the same shaft, and the broken line below R2 indicates an "open" while the solid line above R4 indicates a direct connection.

To use definite values, we will assume that R2 and R4 each have a resistance of 500,000 ohms while C is .02 mfd, C1 is .05 mfd and C2 is .00075 mfd. With the movable arm in the position shown in Figure 6B, condensers C1 and C2 connected in parallel couple the signal to the grid of the next stage, and their values are such that a flat frequency response is obtained.

With the control shaft turned so that the movable contacts are at the lower end of R4, the R2-C circuit which corresponds to that of Figure 5A, is open and thus causes no high frequency attenuation. At the same time, the 500,000 ohms of R4 is in series with coupling condenser C1, thereby increasing the impedance of its circuit. In effect, this reduces the total coupling capacitance and causes greater attenuation of the lower frequencies.

With the control shaft turned so that the movable contacts are at the upper end of R2, condenser C is connected directly from the tube plate to ground and attenuate the high frequencies as explained for Figure 6A. At the same time, condenser C1 is connected directly in parallel with condenser C2 to provide maximum coupling capacitance with minimum low frequency

attenuation. By placing the movable contacts anywhere between the extremities we have explained, the desired degree of attenuation of either the high or low frequencies can be obtained.

BASS COMPENSATION

The characteristics of the human ear are such that when sounds are reproduced at lower than normal volume levels, the low notes appear to be abnormally weak, while when the sound is reproduced at greater than normal level, the low notes appear to be abnormally loud. Therefore, the manual volume control of many radio receivers and audio amplifiers is arranged so that, at low levels, the intensity of the low notes is not reduced as much as that of the higher frequencies. A typical circuit of this type shown in Figure 6C, is commonly referred to as "Bass Compensation".

Starting at point X, the input circuits include potentiometer R, which is tapped toward its lower end, and from the tap, condenser C and resistor R2 are connected in series to ground. The movable contact of the potentiometer, connected directly to the grid of the tube, allows any value of voltage drop across R, from zero to full potential, to be applied across the grid circuit. Due to this action, the potentiometer acts as a volume control.

At low volume settings of the control, the active section of the potentiometer is shunted to ground through a resistance and capacitance combination which as explained for Figure 6A, attenuates the higher frequencies. With the control set in high volume positions, that part of R, in series with C and R2 acts to cause but little attenuation of any frequency.

COMPENSATED VOLUME CONTROL

If, in the arrangement of Figure 6C, an inductance L is connected in series with C and R, to make the circuit resonant at about 1000 cycles where the ear has the greatest sensitivity, then the audio frequencies in this region receive greater attenuation than those at the higher and lower frequencies. This series resonant circuit is a form of an acceptor circuit, sometimes called a "trap", and its effectiveness can be controlled by the value of R. Larger values of R will decrease the attenuating effect of the tuned circuit. As this circuit arrangement gives an apparent boost of both the high and low audio frequencies, it is called a "Compensated Volume Control", in contrast with the bass compensated circuit of Figure 6C.

BASS BOOSTER

The circuits of a simplified "Bass Booster" are shown in Figure 6D, which instead of attenuating the highs in order that

the low notes will be more pronounced, amplifies the low frequencies more than the highs.

Checking the circuits, you will find that the input signal voltage, impressed across the volume control R, has two paths. One is from the upper end of R to the grid of V1 while the other is from the movable contact to the left hand grid of V2, to provide what might be called a dual channel audio system.

Following the signal voltage from the grid of V1, it will appear across the plate load made up of the tuned circuit L-C. From here, it is coupled to potentiometer R2, by condenser C1, and impressed on the right hand grid of V2. As explained in the earlier lessons, a tuned parallel circuit offers maximum impedance at resonance and in a voltage amplifier stage, the gain increases with an increase of plate load impedance. Applying these principles in the circuit of Figure 6, L and C are of values which cause resonance at a low frequency, usually about 50 cycles. At signal frequencies near this value, the plate load impedance and stage gain will be greater than for higher frequencies and thus the lower notes are accentuated or "boosted".

Thus the original signal voltages are impressed directly across the left grid circuit of V2 through the sliding contact of volume control R, while the bass boosted signals are impressed across the right grid circuit of V2 through the sliding contact of the "Bass" control R2. Both plates of V2 connect to a common load, therefore the signal output voltage is a combination of both inputs. In practice, control R is set to provide the desired volume level and control R2 is adjusted to provide the desired bass boost.

P-M SPEAKERS

The basic principles of dynamic speakers have been explained in former lessons, and therefore it is known that their operation does not depend upon the manner of obtaining a strong magnetic field flux. Recently developed magnetic alloys have made it possible to dispense with the field coil and achieve comparable results using Permanent Magnet dynamics, more commonly abbreviated "p-m" speakers.

Figure 7 shows a cross sectional view of a "ring" magnet type of p-m speaker, and as the fundamental parts are associated in the same way, the basic action is like that of an electromagnetic speaker. Other designs make use of the same principle except the core is replaced with a "slug" magnet of the proper strength and the outer shell or case is made of lighter material as it merely provides a magnetic path to the air gap.

As far as tone quality and frequency response are concerned, similar sizes of electro-dynamic and p-m speakers are very nearly identical. The larger sizes of p-m's cost more than the equivalent dynamic, but for remote placement of speakers, the p-m has the advantage of requiring but two conductors from the audio amplifier.

BASS REFLEX PRINCIPLE

To improve the audio frequency range of speakers, one effective method of extending the bass response and reducing cabinet resonance is to cause the back wave from the moving cone to add in phase to the frontal wave moving forward from the face of the cone. In Figure 8 we show the front and sectional side views of a typical Bass Reflex speaker cabinet which operates on this principle and in comparatively small space, provides reproduction similar to that of a speaker mounted on an extremely large flat baffle.

The action can be shown by the sound wave motion of Figure 8B, where the solid semicircles, moving out from the cabinet represent the frontal wave. Similarly, the solid circles inside the cabinet represent the waves which are 180° out of phase with the frontal waves. If the dimension "D" is of the proper length and the port area is in the correct position, the reflected waves shown by dotted half circles will build up a standing wave and be in the correct phase relationship to add to, or reinforce, the frontal wave. The overall action is to extend the bass response and to partially eliminate the cabinet resonant effects that very often occur in open back cabinet baffles.

Another method of improving the bass response is to place the speaker driving unit in an enclosed cabinet and then cut suitable driving unit in an enclosed cabinet and then cut suitable slits, usually in the back, for the escape of the back wave. Some enclosures are made air tight and the inside is lined with absorption material. The stiffness due to air compression in the enclosure acts to aid in radiation of lower frequencies.

DISTRIBUTION ANGLES

You may have noticed the difference in the quality of sound emitted from a given speaker at different angles with respect to the axis of the cone. For example, in Figure 9, a listener at L1 would not hear the same quality of sound as a listener at L2. As high frequency waves, above about 3000 cycles, seem to travel in narrow beams at an angle of about 20° with the axis of the cone, the listener at L1 would hear predominately the lower and middle frequencies, while the listener at L2 would hear the high frequencies also.

The angle of radiation of "beam like" distribution at high frequency widens out as the frequency decreases. However, as it is the higher frequencies in speech and music which give understandability and brilliance, it is desirable to diffuse or conduct the higher notes to all parts of the listening area.

To accomplish more uniform distribution of sound at all frequencies, several methods are used, one of which is shown by Figure 10. Here, in the top view at A and side view at B, we show vanes which, if you visualize Figure 10A replacing Figure 9, will cause the high frequency radiation to be spread. Therefore, with the vane diffusers, listeners at L1 and L2 of Figure 9 would hear very nearly the same proportion of high and low frequencies.

The vanes of Figure 10 cause spreading of the high frequencies only in the horizontal plane. Normally this is sufficient for room reproduction of audio programs, but should a greater spread or distribution be desired, vanes can be placed horizontally to provide vertical distribution. Figure 11A illustrates the double vane idea carried a little further in that the vanes now take the shape of a multicellular horn. Loudspeakers of this type are primarily for use in Public Address installations such as theaters, skating rinks, churches, etc.

In order to achieve 360° sound distribution, the plan of Figure 11B can be used. The entire unit is suspended from a suitable support and because of the shape of the upper and lower deflecting plates, as indicated by the arrows, sound waves will spread in a 360° radiation angle with the vertical axis of the speaker assembly. The driving unit is contained in a suitable metallic housing which in turn is attached to the polished deflecting plates.

TWO WAY SYSTEMS

Earlier in this lesson, we explained the bass reflect principle to increase the bass response of a speaker and although not mentioned previously, we show a high frequency speaker unit in the cabinet enclosure of Figure 8.

It has been found that a single cone speaker does not adequately cover the extended frequency range desirable in high quality reproductions. If frequencies from about 40 to 15,000 cycles are desired, this can be accomplished best by using two separate speakers -- a "woofer" for low frequencies, and a "tweeter" for high frequencies, to provide a two way system. The woofer can then be constructed with a rather large cone area and heavy voice coil in order to develop the necessary power to radiate low notes. For high frequencies, the cone area can be small

and the voice coil can be light to move more rapidly and with smaller power requirements.

To direct the electrical currents of high and low audio frequencies into their respective channels, a frequency divider filter system is needed.

As shown in Figure 12, in simplified form the filter consists of an inductor, L1 and condenser C2 connected in series across the amplifier output. Due to the relationship between frequency and reactance, with a low frequency amplifier output voltage, most of the drop will appear across the condenser while, with a high frequency amplifier output voltage most of the drop will appear across the inductor. The value of which both reactances are equal is known as the "cross-over" frequency.

With the high frequency speaker connected across the inductor and the low frequency speaker connected across the condenser, the desired range of frequencies is impressed across each. Condenser C1 and inductor L2 act as an added high pass filter section in the high frequency speaker circuit to provide a sharper cut-off of the lower frequencies.

One model of a two way speaker system uses the plan of supporting the high frequency speaker in the cone cavity of the large speaker. For example, imagine the high frequency unit of Figure 8B properly supported in the center of the large cone area of the speaker just below. Such a method simplifies cabinet construction and, at the same time, provides some sound distribution.

Another two way speaker design system provides for the mounting of the high frequency unit on the rear of the woofer cone driver, and the high frequency sound is fed through a passage in the center of the woofer pole assembly. In effect, this passage is a horn, the cross section area of which expands for proper distribution of the sound waves.

EFFICIENCY

Very little has been said about the efficiency of a speaker. That is, the conversion of electrical power to acoustical energy. It is very difficult to obtain the necessary data for determining speaker efficiencies, and such calculation is usually performed only by the manufacturers in the research laboratories. However, we want to give you an idea of the relative capabilities of different speakers, and it may surprise you to learn that loudspeakers are not very efficient. The efficiency of a loudspeaker may be defined as the ratio of the useful power output (acoustical) to the electrical input (watts) expressed as a percentage.

In the following table the values are general deductions based on the use of a 400 cycle test note which approximates the most predominate frequencies in musical reproductions.

SPEAKER TYPE	EFFICIENCY
Small Cabinet	5 to 10%
Large Baffle	5 to 10%
Directional Baffle	7 to 20%
Projection (Horn)	10 to 30%

SELECTION OF SPEAKERS

The size and type of speaker system in an installation is governed almost entirely by the size, type and location of the audience to be covered, the type of sound to be reproduced and the psychological reaction desired. Therefore each installation must be analyzed before logical conclusions can be reached.

For an indoor installation considered quiet, such as the small office, theater, class room, hospital, church and funeral parlor, about 5000 square feet of space can be covered per 5 watts of audio power. This is considered only as a "rule of the thumb", and is subject to considerable variation with each installation.

More definite approximations of required power are shown by the graph of Figure 14. The upper limits of shaded area should be used for locations with high noise level, or where maximum fidelity of music is desired. Noise decreases the intelligibility of speech so the required level of the reproduced voice must be boosted to overcome this effect. For example, if the volume of the enclosure is 40,000 cubic feet, reading from the graph, the required power is approximately 5 watts for quiet installation. For high noise installations, the required power is approximately 10 watts. It will be noticed that the power variation between noisy and quiet installations increases with the volume.

For outdoor installations, the required audio power is roughly 5 watts per thousand square feet. Of course, the surrounding outdoor noise level and the indoor noise level and acoustics of a room has considerable bearing on the final choice of the speakers. In order to meet emergencies it is advisable to provide a system with a reserve power up to three or four times the estimated requirement.

To create the desirable illusion that all sound is heard from its point of origin, the speakers should be located as close

to the source as possible. For outdoor installations this requires higher powered speakers than if the sound were reproduced at various points throughout the audience, but eliminates the distracting effect of watching a performer in front and hearing his voice from behind or overhead. Whenever it is necessary to distribute sound other than from its point of origin, it is advisable to operate the system at as low a level as is consistent with intelligibility.

Where the reproduction of both voice and music is required, a large horn, baffle or suitable speaker enclosure is of importance in order to reproduce the low frequencies. In this instance a speaker arrangement like that described for Figures 8 and 12 would be preferred. In place of the high frequency cone speaker, the multicellular speaker like that of Figure 11A could be used. By proper selection of the "cross over" frequency, the cellular horn could be used for reproducing the voice frequencies, and both speakers for full range sound.

Looking down, Figure 13 shows the relative angles and depths of audience coverages with three popular arrangements of speaker baffles and it will be noticed that the greatest intensity of sound is along the axis of the speakers. The flat baffle of Figure 13A is in most common use, and it spreads the sound over the widest possible area. The projector type baffle of Figure 13B is suitable for a more concentrated distribution of sound as it provides a sharper angle of coverage with penetrating tones for forcing sound under balconies, and out over areas like athletic fields. When it is desired to concentrate the sound into a comparatively narrow beam, the trumpet arrangement of Figure 13C is most efficient. This type is used more extensively for stadiums, grandstands, belfry chime systems, and in noisy locations where highly directional characteristics are desired.

One factor to keep in mind is the distance over which the sound must be projected from given speaker system. Sound energy diminishes approximately as the square of the distance from the source. For example, if a speaker system provides the desired sound at a distance of 50 feet with an input of 5 watts, at ten times the distance, 500 feet, 10 x 10 x 5 or 500 watts will be required. Therefore, both the speakers and amplifier system must have ample reserve power.

PLACEMENT OF SPEAKERS

In most installations, the speakers are placed above and near the microphone in order to allow their sound and that of the performer to reach the audience at the same time. If the speaker is remote from the source and a listener near the

performer, the waves arrive at slight differences in time, and result in indistinct sound.

In connection with the problem of obtaining most uniform sound distribution in front of a horn loudspeaker, Figure 15 shows a method by which it can be made reasonably uniform on a path between the loudspeaker and a point at a given distance directly in front. This requires that the speaker be mounted at a height "h" and directed downward at a tilt angle "s" so that listeners along the floor line, CBA, hear the same volume of sound.

For example, assume the horn speaker of Figure 15 provides a distribution pattern similar to that of Figure 13C so that at 30° off the axis the sound pressure of some selected frequency is reduced 50%. Then, a listener at B should hear but half the volume as listener at A long the axis. However, with "h" and "s" at proper value, the listener at B is but half the distance from the horn, D/2, as the listener at A, and therefore both hear the same volume of sound.

Although there are many configurations of auditoriums, halls, ballrooms, etc., Figure 16 is a typical floor plan of two common sound reinforcement arrangements. If the auditorium of Figure 16A has dimensions of 110 feet in width by 45 feet in length with a ceiling up to 20 feet, its volume is 99,000 cubic feet. Considering the noise level to be reasonably low, and determining the required power on the basis of 100,000 cubic feet, the curve of Figure 14 shows a minimum requirement of 30 watts. Therefore, one 30-40 watt amplifier would be recommended with four 10 watt speakers arranged as shown.

If the ballroom of Figure 16B is 100 feet in length by 40 feet wide with a ceiling height of 15 feet, its volume is 60,000 cubic feet. Considering the noise level to be high, the curve of Figure 14 shows a required power of approximately 17 watts. Therefore, one 15-20 watt amplifier would be recommended, with two 10 watt baffle type speakers placed as shown.

In auditoriums smaller than those indicated for the above recommendations, only one speaker may be needed, and it should be placed above the microphone, and slightly forward in the direction of the audience.

The placement of speakers for outdoor stadiums requires a different plan to arrive at adequate distribution. In general, projectors or trumpets are placed to direct the sound toward the stands. Depending on the size of the stand, one speaker may serve for one or two sections. The speaker should be placed 25 or 50 feet in front of the stand, and high enough to tilt slightly toward the main body of the audience.

PREVENTION OF FEEDBACK

The placement of the speakers has been generally covered only with respect to area, but it is also necessary for the loudspeaker to be placed in a position to minimize acoustic feedback. Feedback is caused by sound, produced by the speakers which returns to the microphone and builds up to a "howl", which destroys all other sound except itself. To prevent feedback, the speakers are placed or positioned so that the sound does not return to the microphone with any degree of strength.

When the possibility of feedback still exists, although the speakers are suitably placed, it may be necessary to use unidirectional microphones in order to minimize the action.

Another corrective measure for permanent installations is that of treating the room acoustically. This is recommended only when the reverberation is unusually high, and the treatment is an improvement in "acoustical presence".

IMPEDANCE MATCHING

In the output circuit of an audio amplifier, the plate circuit of the output tubes can be considered as the generator or source and the voice coils of the speakers as the load. Like any other a-c system, the maximum transfer of energy occurs when the impedance of the source equals that of the load. In this case the impedance of the source can be found for any particular type of output tube by checking the recommended value of "load resistance" as given in a table of tube characteristics. For the common types of tubes, the recommended load resistances vary from about 1000 ohms up to 30,000 ohms and above.

Measured usually at 400 cycles, the voice coil impedance is much lower than the recommended value of load resistance, therefore an impedance matching output transformer is installed as a coupling unit. Connected in series with the plate of the output tube, the transformer primarily becomes the actual load but its effective value of impedance is controlled by the turns ratio and voice coil impedance.

For example, assume 6 watts of power developed in a 3200 ohm load resistance is to be coupled to an 8 ohm speaker voice coil. For the load resistance:

$$E = \sqrt{WR} = \sqrt{6 \times 3200} = \sqrt{19200} = 140 \text{ volts (approx)}$$

$$I = W/E = 6/140 = .0428 \text{ amp} = 43 \text{ ma (approx)}$$

For the voice coil:

$$E = \sqrt{WR} = \sqrt{6 \times 8} = \sqrt{48} = 6.9 = 7 \text{ volts (approx)}$$

$$I = W/E = 6/7 = .857 \text{ amp} = 860 \text{ ma (approx)}$$

The voltage ratio is 140/7 = 20 and assuming a perfect transformer, it should have a step-down turns ratio of 20 to 1. Under these conditions, for equal power, the secondary current will be 20 times that in the primary. Thus, by transformer action, in this example, the 8 ohms impedance of the voice coil appears, or is reflected, in the plate circuit as 3200 ohms. The impedance ratio is 3200/8 = 400 and the square root of 400 = 20 which is the turns ratio. As this relationship holds true in all similar cases, as a general

$$N = \sqrt{\frac{Z_p}{Z_s}}$$

When N = turns ratio of the transformer
Z_p = effective or required plate load impedance, in ohms
Z_s = impedance at secondary load, in ohms

In practice, a transformer with a turns ratio for exact impedance match is not always available and in some cases a mismatch as high as 25% is not objectionable. However, for greatest efficiency and lowest distortion, the mismatch should provide an effective plate load impedance that is greater than the recommended value.

TRANSMISSION LINES

In addition to the impedance match of the output transformer, two other factors are important when connecting speakers to an amplifier. First, the power loss due to line resistance and second, the high frequency loss due to the line capacitance. In general, if the distance between the amplifier and speaker does not exceed 30-35 feet, the impedance of the connecting line is not important and the most convenient value may be used. For greater distances, it is necessary to take the resistance and capacitance of the leads into consideration.

If a line is operated at the voice coil impedance, its ohmic resistance should not exceed about 15% of that of the voice coil, in order to limit the line power loss to approximately 15% of the power delivered to the speaker.

When the line power loss exceeds 15% it is preferable to use a transmission line (500-600 ohms) with efficient impedance matching line transformers at each end. The line loss, due

to capacitance shunting effects, increases with the impedance of the line. That is, at low values of line impedance (voice coil) the shunting effect is very nearly negligible. However, the shunting capacitance becomes an important factor when the impedance of the line is high particularly when it is in the input circuit of a sound system.

SPEAKER PHASING

For installations with more than one speaker, it is important to connect them so that all the cones or diaphragms move in the same direction at the same instant. Unless this is done the total output will be reduced materially, because sound waves from one speaker will tend to cancel those from another. To obtain the desired conditions, it is necessary to "phase" the speakers by checking the direction of cone movement with the polarity of an applied voltage.

To make the test the speaker must be in operating condition therefore, the fields of dynamic speakers must be excited and humbucking coils shorted temporarily. A standard #6 dry cell provides a low voltage d-c sound and, for convenience a short length of wire is connected to each of its terminals. The dry cell is then connected momentarily as is each voice coil in turn and its cone is matched to check the direction of its initial movement or "jump" as the test circuit is closed. Usually the cell connections are reversed if necessary so that all cones jump in the same direction, in or out, and the voice coil terminal connected to the dry cell positive is marked for identification.

For parallel operation, all the marked terminals are connected to one side of the line and all the remaining terminals to the other side of the line. For series operation, marked and un-marked terminals are connected like terminals in the usual manner for series circuits.

If it is desired to phase several speakers each having its own transformer attached, the same procedure is followed except that a battery of about 22.5 volts is connected across the primary of the transformer. Voice coil leads must be attached to the transformer secondary, and bucking coils shorted out temporarily. Figure 17 shows the arrangement of this phasing test whereby the d-c voltage is "touched" to lines x and y, while noting the direction the cone moves. Any color coding, terminal identification or wire position is then recorded so that the desired operating conditions can be obtained.

POWER SUPPLIES FOR MOBILE P-A SYSTEMS

Previous explanations of p-a systems have assumed that the conventional 110-120 a-c 60 cycle source of power has been

available for the operation of the equipment. However, many instances require that sound systems be mobile in nature and under such conditions the source of power may be ~~(a) Storage~~ batteries or (b) Prime mover (gas engine) type a-c generators.

Operation from d-c sources is accomplished by two general types of power supply equipment. One incorporates a vibrator and operates on the principles explained for auto radios while the other employs rotating machines, known commonly as Dynamotors and Rotary Converters.

DYNAMOTOR

As shown in the simplified sketch of Figure 18, a dynamotor consists of a low voltage d-c motor, designed to operate on a low voltage storage battery supply. The armature carries a second winding, with a comparatively large number of turns, brought out to a second commutator and set of brushes.

Revolving with the motor armature, the turns of this second winding cut the magnetic flux developed by the shunt field coils and thus operate exactly as the armature of a generator. The commutator and brushes provide a d-c output while the large number of turns of the armature winding produce the required high voltage.

You can think of this arrangement as a low voltage d-c motor driving a high voltage d-c generator but, to save weight and space, both units operate in the same frame and magnetic circuits. The overall effect is much like that of a step-up transformer except that a low voltage d-c input provides a high voltage d-c output.

ROTARY CONVERTERS

As shown in the sketch of Figure 19, the rotary converter is mechanically similar to the dynamotor with a shunt wound motor operating on a d-c supply. The armature has but one winding and, instead of a second commutator, a pair of insulated slip rings are mounted on the shaft. The rings are connected separately to diametrically opposite points on the commutator and contact stationary brushes connected to the external output circuit. As the armature rotates, the rings connect alternately to the d-c brushes and thus there is a-c across them.

The main function of the motor is to revolve the armature so that the commutator and its brushes will act as a reversing switch between the d-c input and a-c output. Thus the frequency of the a-c output depends on the number of field poles

and the speed or rpm of the armature. Ignoring any losses, the d-c input voltage equals the maximum value of a-c output voltage so that an ordinary a-c voltmeter connected across the slip rings will read the effective a-c value which is equal to .7 of the maximum. With a 12 volt d-c input the output would be 12 x .7 = 8.4 volts a-c effective, but by means of transformers, this low voltage a-c can be stepped up to the desired values.

In mobile sound systems where the required power input is quite high, and time of service is longer than can be supplied by storage batteries, it is feasible to drive an a-c generator with a small gasoline engine.

SOUND MOTION PICTURE SYSTEMS

The sound system used in a motion picture theater consists of a photo-electric cell pickup device, audio frequency amplifiers, and a set of loudspeakers. The pickup device serves to convert the light variations produced by the film "sound track" into corresponding electrical variations. The latter are then amplified and fed to the speakers which are located behind the screen on which the picture is projected.

A photo-electric cell is similar to a radio tube in appearance and contains two elements, one of which the cathode is coated with a light-sensitive material such as Potassium or Caesium and the other serves as the anode or plate. The cathode coating material emits electrons when exposed to light and if the photo-electric cell (pec) is connected into a closed circuit, it will produce current the amplitude of which will be proportional to the intensity of the light striking the cell.

The sound track on a movie film consists usually of a strip along one edge of the film. This strip varies in transparency from point to point at the rate of and in proportion to the amplitude of the sound signals which accompany the picture.

The sound track transparency-variations are changed to audio frequency electrical variations by causing a steady light to strike one side of the track, while a pec, placed on the other side of the film, will be affected more or less from instant to instant in accordance with the varying amounts of light transmitted through the sound track. The pec current, varying at audio frequencies, passes through a series resistor and the a-f voltage drop thereby produced is applied to the input of the amplifier system. From here on the circuits are much like those of the p-a systems described in this lesson.

RADIO INPUT

As previously mentioned, it is possible to connect the audio output of a radio second detector or "tuner" into the input of the p-a system and thus permit any available radio program to be heard by a large group of people. Since the output of the detector will usually be on the order of a few volts, the p-a system phono input should be used. Some commercial audio amplifiers include an input connection which is specifically designated as the "radio input".

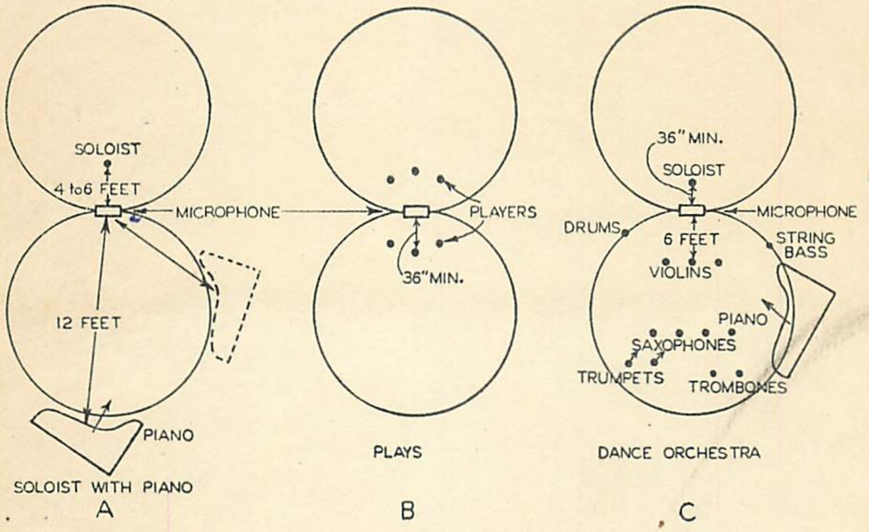

FIGURE 1

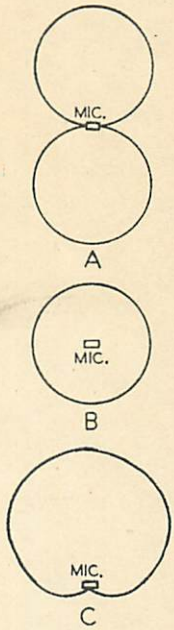

FIGURE 2

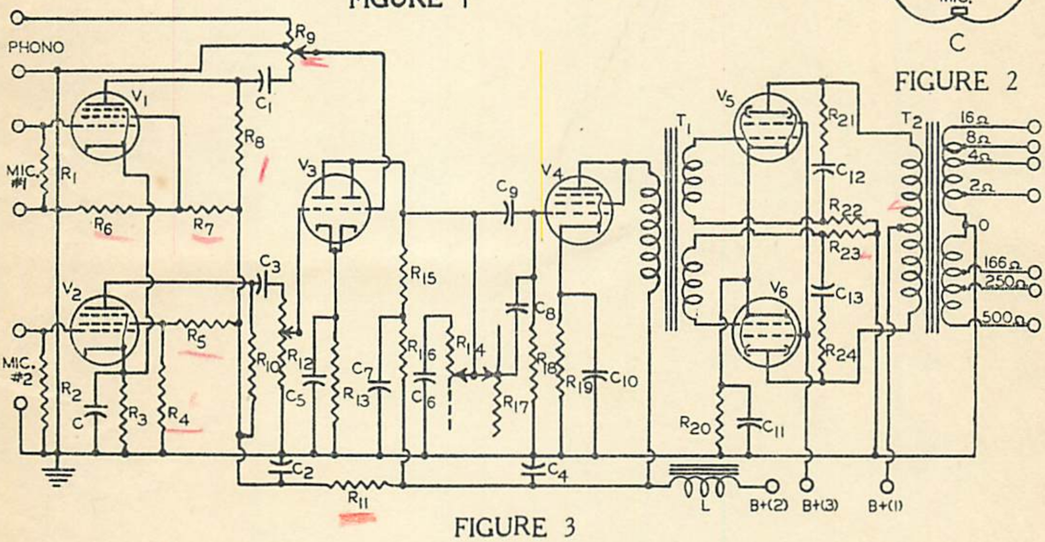

FIGURE 3

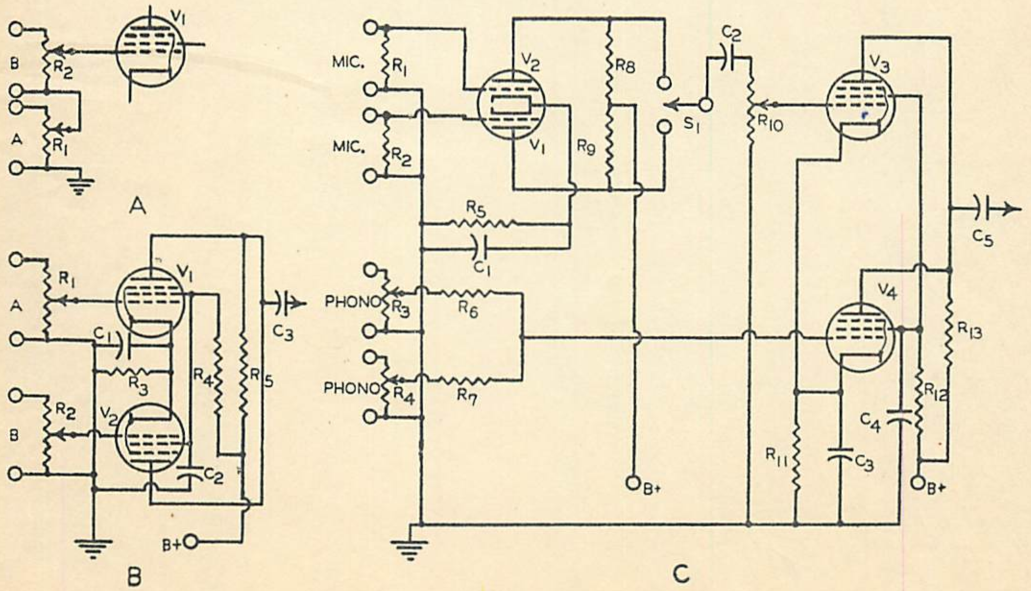

FIGURE 4

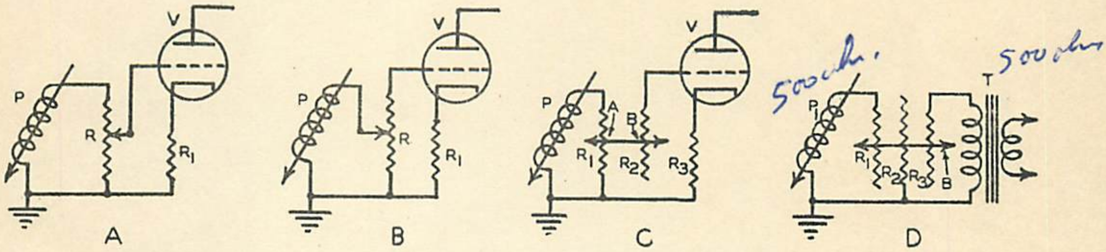

FIGURE 5

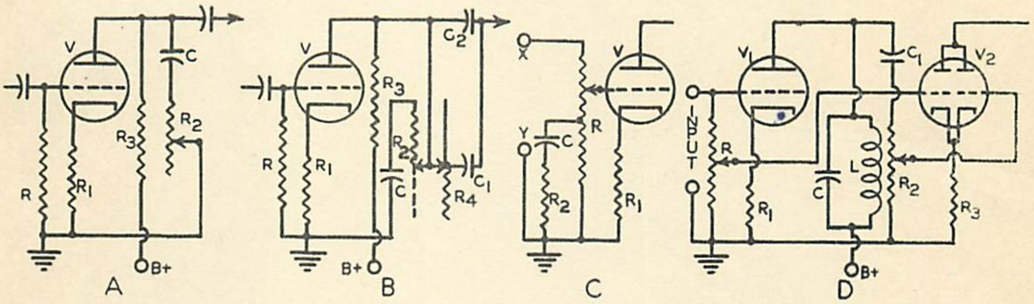

FIGURE 6

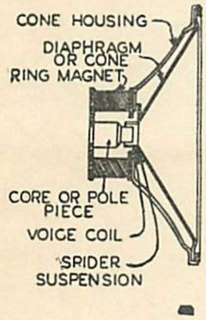

FIGURE 7

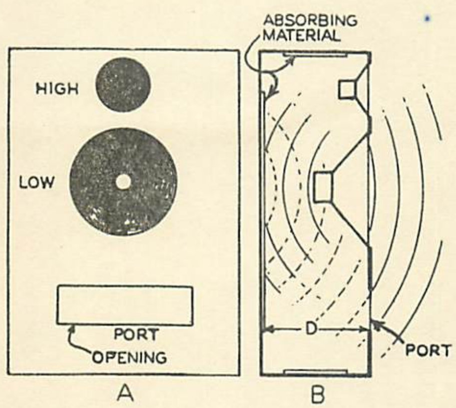

FIGURE 8

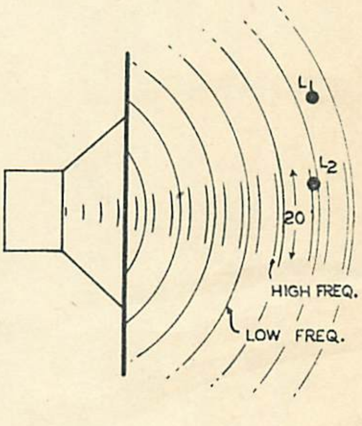

FIGURE 9

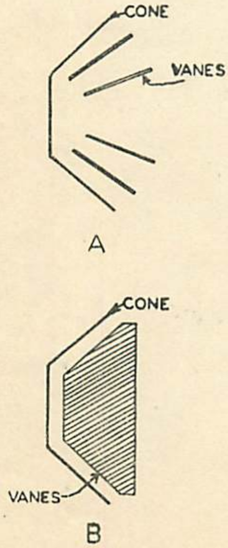

FIGURE 10

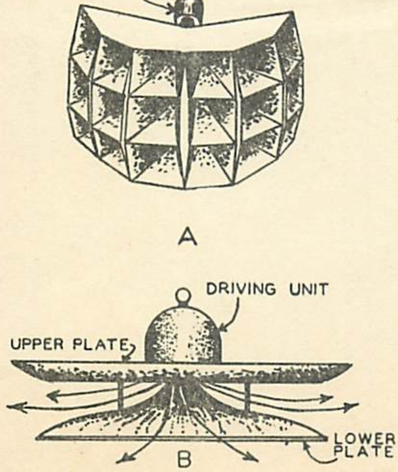

FIGURE 11

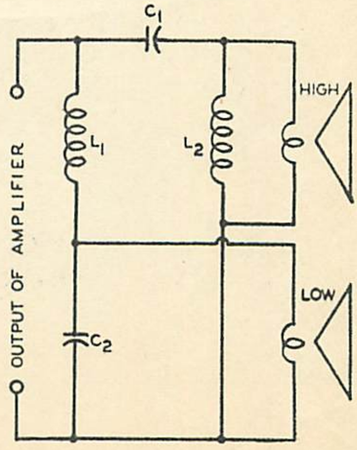

FIGURE 12

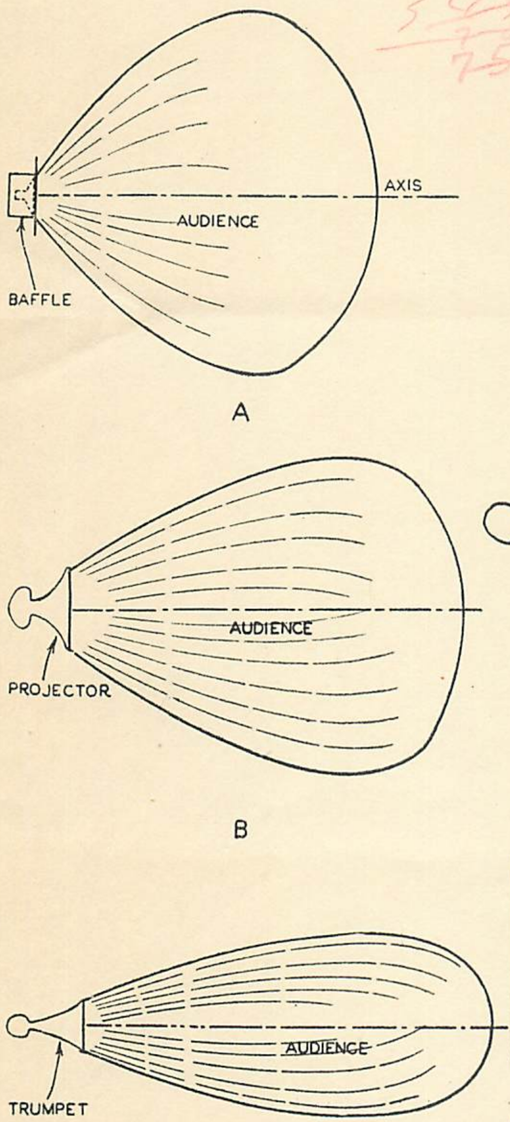

FIGURE 13

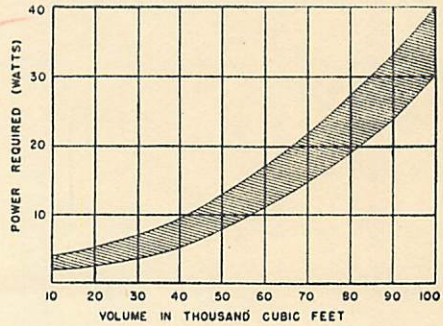

FIGURE 14

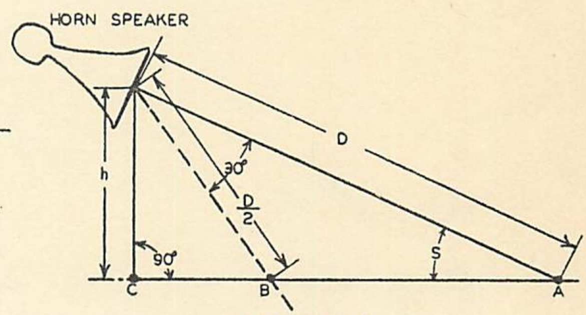

FIGURE 15

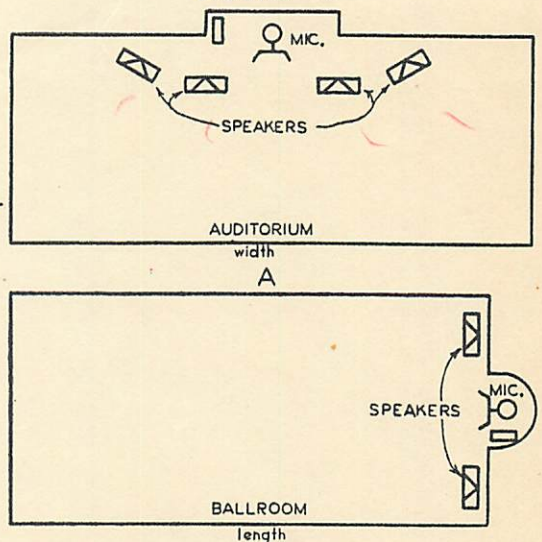

FIGURE 16

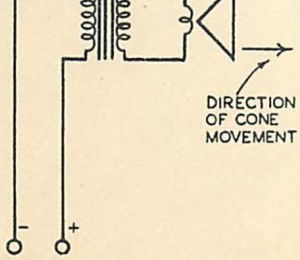

FIGURE 17

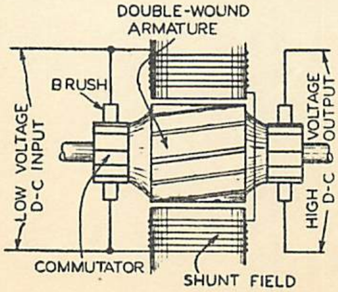

FIGURE 18

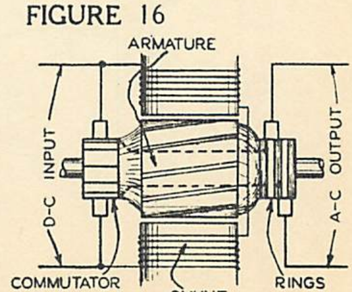

FIGURE 19